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Convergence in Local Telephone Networks

Softswitch and Packet Voice

CopperCom

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Executive Summary

This paper is the third in a series of technical publications on the topic of voice over DSL. It is intended for network planners and technologists who are responsible for the planning and design of local broadband and voice access networks. The paper follows previous publications that covered the market requirements for voice over DSL, and discussed practical aspects of implementing voice over DSL networks. In this paper, the broader question of network evolution is discussed, with a focus on softswitch technology and packet voice and their roles in the migration to a fully converged local network

About the Author

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1 Introduction

The unbundling of network elements, a key aspect of the de-regulation of telecoms networks that is now taking place all around the world, has finally opened up the local telephone services market to competition. Competitive service providers have been able to enter the market and win a substantial share, particularly for medium-sized business customers, those with 100 employees or more. Recently, packet voice solutions that exploit Digital Subscriber Line (DSL) transmission technology have dramatically improved the business model for deploying voice over unbundled loops. Voice-over-DSL (VoDSL) solutions are enabling competitive service providers for the first time profitably to address small business and upscale residential customers, while incumbents plan to exploit VoDSL both as a defensive strategy and to address copper pair shortage problems.

VoDSL technology undoubtedly provides a major boost to competition in local telephony markets by greatly reducing the cost of local access. But this use of packet voice within the access network has done nothing to change the switching infrastructure for local telephony, which continues to be based on traditional circuit-based local exchange switches. The high cost and large scale of this equipment is a major deterrent to any service provider that wishes to enter new markets, particularly in second and third tier cities. Furthermore, service providers are limited by the capabilities of the switch to offering voice services and calling features that are identical to those provided by the incumbent. As a result, service providers are forced to compete almost solely on price, and this is unlikely to provide a sound basis for long-term business viability.

A technology known as "softswitch" has the potential radically to change the landscape of local exchange switching. Properly applied in the local network environment, softswitch can deliver an irresistible combination of dramatically lower equipment cost together with greatly enhanced calling feature functionality. This paper will show how softswitch will both revolutionize local telephone services, and promote the graceful evolution of access networks from circuit-based to packet-based – a key step in the migration to end-to-end packet voice.

2 Local Access Today

The local access network today comprises transmission facilities that link customer premises equipment and networks to local exchange switches. Traditionally, the local access network has included a combination of analog local loops and circuit-based digital transmission facilities.

2.1 Incumbent Local Access

Customers with fewer than 12 to 16 lines are generally served by means of analog local loops. Where the customer is located within about 20,000 feet from a central office, the local loops run directly from the customer premises to the central office. Customers located beyond the reach of direct analog connections are served by outside plant cabinets that contain Digital Loop Carrier (DLC) systems. A DLC connects analog loops from customer premises to digital transmission facilities such as repeatered T1 or E1 lines that link the DLCs back to the central office.

Customers with more than 16 to 20 lines may be served more economically by means of direct digital transmission facilities into the customer premises, most often T1 or E1 lines. These facilities may be connected directly to digital PBXs or key systems, or they may be dropped off as analog lines by means of a conversion device known as a channel bank.

2.2 Competitive Local Access

Access to unbundled network elements including copper loops, and the ability to collocate transmission equipment in the Central Offices of incumbents, have enabled competitive service providers to offer local phone services using a variety of traditional, i.e. circuit-based access techniques.

For business subscribers requiring more than 20 lines or so, competitive service providers can lease a T1 or E1 access line from the incumbent. These lines may run direct from the customer premises to the competitive service provider's local switching end office, or they may be groomed into higher speed facilities such as T3 or E3 lines using collocated multiplexing equipment.

T1 and E1 access lines generally do not provide a cost-effective solution for subscribers requiring 16 lines or less. To date, competitive service providers have had to serve these subscribers using an individual unbundled loop for each line, with Digital Loop Carrier (DLC) systems in collocation to concentrate the traffic onto digital facilities for transport to their local switching end offices.

Emerging Voice-over-DSL (VoDSL) solutions provide a far more economical solution for subscribers requiring 16 lines or less. VoDSL leverages the high-speed transmission and packet transport of the DSL access network to deliver multiple lines of telephony plus high-speed data over a single unbundled loop. VoDSL has the further advantage that voice and data traffic can share common ATM-based local network facilities for traffic concentration and backhaul to the competitive service provider's local switching end office. VoDSL can provide a cost-effective solution for competitive local access down to as few as 4 lines of telephony.

2.3 Access Network Evolution

The emergence of the packet-based broadband transmission technologies that we know collectively as DSL is having a profound impact on the evolution of the access network. From a pure transmission standpoint, DSL has been around for quite a number of years – ISDN is, technically speaking, a variety of DSL. But ISDN and other related variants of DSL such as HDSL made use of traditional time-division multiplexing techniques, which limited their appeal. The very rapid success of today's DSL offerings is based on their exclusive use of packet-based transport, which provides a very cost-effective solution for data applications such as Internet access.

VoDSL leverages the economies of broadband packet transmission in the access network and enables service providers to deliver integrated voice and data services over a common infrastructure. Many industry observers are now coming to the conclusion that the future of the telephony access network lies with packet voice over DSL.

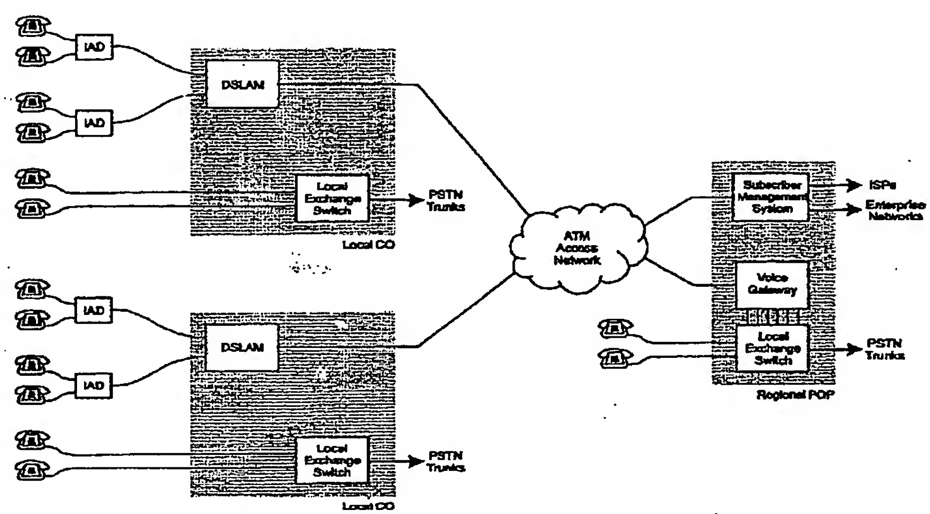

Figure 1 : VoDSL Architecture for Competitive Service Providers

3 The Impact of Packet Voice

We have already identified the impact of packet voice in the access network, where the economics of leveraging DSL to transport voice are quite compelling. Now let us turn to the question of packet voice in the trunk network, and the prospects for end-to-end packet voice.

3.1 Historical Perspective

The packet voice revolution has slowly but steadily been gaining momentum over the last few years. The earliest deployments of packet voice were driven by enterprises using wide-area data networks to carry intra-enterprise voice traffic as a cost-saving measure. The success of this approach was, however, short-lived, because long-distance service providers brought in dramatically lower prices for virtual private voice networks, all but eliminating the cost advantages of packet voice on private data networks.

Internet telephony has been another key area of application for packet voice. The flat rate access charges for Internet communications offer opportunities for major cost savings on long distance and international calls. However, the relatively poor performance of the Internet in supporting real-time voice communications has limited its success as an alternative to the traditional PSTN, apart from its possible use to carry fax traffic, which is largely insensitive to transmission delays.

3.2 Outlook

Despite the patchy success of packet voice to date, market interest in this area remains very high. The primary reason for this interest is the dramatic growth of IP traffic relative to voice traffic in networks all over the world. The bandwidth consumed by voice traffic in the PSTN is growing much more slowly than the rate at which bandwidth consumption is growing in the Internet. If Internet traffic continues to grow at the same rapid pace, in a few years time the total volume of voice traffic worldwide will be dwarfed by the volume of IP traffic, most of which is Internet-related.

Growth in the volume of IP traffic is driving demand for extremely rapid advances in both transmission and packet switching technologies. Dense Wavelength Division Multiplexing (DWDM) techniques are squeezing multi-Gigabit capacities out of individual optical fibers, while core network routers are matching this growth in transmission capacity with multi-Gigabit routing capabilities. As a result of these continuing advances, transmission and switching costs for IP traffic are falling fast. It is not hard to conclude that voice transmission and switching costs would also fall fast if voice were carried in packet form. Since there is no comparable innovation taking place in the circuit-switching world, one may also conclude that voice transmission and switching costs will not fall anything like so quickly if voice stays on the traditional circuit-switched network. This vision is driving the rapid developments that are now taking place in packet voice – one of which, of course, is the emergence of VoDSL solutions for packet voice access.

It is tempting to conclude that, in a world dominated by IP, voice will ride almost for free on this high-capacity IP infrastructure. But the requirements for voice transport are fundamentally different to those for data, and it is unclear at this point how a pure IP-based solution will be able to provide guaranteed delivery of voice packet streams with acceptable end-to-end transmission time on a global scale. This is a major concern to service providers, since it is voice traffic that continues to dominate both their revenues and their margins. As a result, most of the IP traffic in these high-capacity packet networks is actually being carried over ATM, since ATM offers common packet-based transport facilities for both voice and data while providing the Quality of Service (QoS) guarantees that are so important to voice.

As we look forward, the economic case for applying packet-switching techniques to voice are very clear. What is less clear right now is which packet switching technique will be dominant in the voice world in the long term. The proponents of IP-based voice may argue for the elimination of the ATM layer altogether, but until IP has demonstrated a clear ability to provide a scalable and economically attractive solution that protects the quality of voice transmissions, the safe choice for voice will continue to be ATM.

4 Challenges for Local Voice Service Providers

Enterprise customers and medium and large businesses enjoy the benefits of a highly competitive market for telecoms services today. The volume of traffic that is generated by these customers justifies the use of dedicated circuits from customer premises to multiple local and long distance networks, and the PBXs that are used by these customers provide call routing intelligence that directs traffic on a call-by-call basis to the most appropriate carrier. The telephony services purchased by these customers typically relate to the basic transport of voice between multiple enterprise locations and the connection of voice traffic into various public networks.

The small business or upscale residential customer with fewer than 16 lines does not have the advantage of multiple competitive service offerings. The economics of access do not support direct connections from these kinds of customers into long distance networks. They are therefore dependent on a single incumbent service provider for local dial tone and for access to all other services.

These market segments generate the bulk of profits for incumbent local service providers. Service providers earn large amounts of revenue from per-minute charges on local calls, from origination and termination of long distance calls, and from usage of custom calling features and other local services such as voicemail. And since there has been no effective competition in these market segments, tariffs are high relative to costs, and margins are therefore very healthy.

Today, all local voice service is delivered from traditional circuit-based local exchange switches, largely because no other solution exists. This represents a major barrier to progress in local voice services, for three key reasons.

4.1 Cost of Local Exchange Switching Solutions

The market for local exchange switches is dominated by a few large players who have built highly profitable businesses serving the needs of incumbent local service providers. Local exchange switches from these vendors are optimized to suit end offices that serve many tens of thousands of lines. While the scalability of these devices is not in doubt, high common equipment costs make these solutions wholly unsuited for smaller deployments where only a few thousand lines are needed. An entry-level local exchange switch typically costs of the order of $5 million, an amount that seems certain to deter competitive service providers from entering any but the largest markets.

Some competitive service providers have succeeded in entering second and third tier markets with traditional local exchange switches, by installing a single switch to serve multiple cities. The switch is logically sub-divided into multiple local exchange codes, one or more for each city, and connected by trunking facilities to local access networks in each city. However, while this approach does deliver lower switch costs per served line, any savings that are made in switching hardwarew must be set off against the very substantial additional transmission costs of backhauling traffic from each city to a central switch location, and then sending all local call traffic back to the city from which it originated.

If local exchange switching solutions were available at much lower entry-level cost than traditional circuit switches, competition in local voice service would be stimulated, and customers would benefit from a wider choice of service providers and lower tariffs.

4.2 Lack of Service Differentiation

Local exchange switches all offer broadly the same set of features for custom calling services such as call waiting, call forwarding, caller id, selective call blocking and so on. Most of these features have been available for a number of years, and the delivery of completely new features is now relatively rare. This is partly because it is so expensive to develop and test new features, and partly because the feature set available today comprises virtually every feature it is possible to imagine a subscriber being able to program from a telephone keypad.

Since both incumbents and competitive local service providers are dependent today on the same limited set of local exchange switch products, they are constrained to offering exactly the same set of services. In these circumstances, the only way for a competitive service provider to win customers is to offer lower rates.

Differentiation solely on price has not proven to be a good long-term business strategy in the telecoms business. If local exchange switching solutions were available on which genuinely new and attractive features could be offered, then service providers would have a chance to differentiate on services rather than on price. This could offer a much brighter outlook both for margins and for customer retention.

4.3 Barrier to Network Migration

Local exchange switches are all based on circuit switching techniques. Within the switch fabric, voice traffic is represented as 64 kbps streams, while at the input and output ports of the switch, the 64 kbps streams are time-division multiplexed into higher-speed digital facilities. The intelligence of the switch that performs call routing and feature processing is tightly integrated with the circuit-switching fabric.

As we have seen above, the economic advantages of packet voice are driving the evolution of both access and core voice networks away from circuit switching and towards packet. As packet voice becomes widely adopted for both access and core networking, the traditional local exchange switch represents an island of circuit switching that connects these two packet voice networks. The packet-to-circuit conversion that must be carried out at both input and output of the local exchange switch introduces undesirable additional cost and transmission delay into the voice path.

If a local exchange switching solution were available that were capable of delivering local voice services and custom calling features directly over a packet switching infrastructure, then unnecessary packet-to-circuit conversions could be avoided. This has the dual effects of reducing cost and improving quality, and moves the voice network a major step closer to the ultimate goal, which is homogeneous end-to-end packet voice.

5 Introduction to Softswitch

Softswitch is the generic name for a new approach to telephony switching that has the potential to address all the shortcomings of traditional local exchange switches identified above. In this section we explain the concept of softswitching, and in the following section we demonstrate how softswitch solutions can lower the cost of local exchange switching, offer the means to create differentiated local telephony services, and ease the migration of networks to support packet voice end-to-end.

5.1 The Softswitch Concept

By far the most complex part of a local exchange switch is the software that controls call processing. This software has to make call routing decisions and implement the call processing logic for hundreds of custom calling features. Today's local exchange switches run this software on proprietary processors that are tightly integrated with the physical circuit switching hardware itself.

In the future, we will want to deliver local telephony over a pure packet-based infrastructure, although the migration path to end-to-end packet will require us to work with a hybrid network handling both packet and circuit voice for many years to come. But the inability of existing local exchange switches to deal directly with packet voice traffic is a major barrier to packet voice migration.

As one possible solution to this, we can imagine creating a hybrid device that can switch voice in both packet and circuit formats, with all the necessary call processing software integrated into this switch. While this approach may help us address the question of migration, it does not necessarily lower the cost of local exchange switching or improve the prospect for differentiated local voice services.

The telecoms industry appears to have reached broad consensus that the best answer lies in separating the call processing function from the physical switching function, and connecting the two via a standard protocol. In softswitch terminology, the physical switching function is performed by a Media Gateway, while the call processing logic resides in a Media Gateway Controller.

There are a number of reasons why this separation of functionality is believed to be the best approach:

- It opens the way for smaller and more agile players who specialize in call processing software and in packet switching hardware respectively to make an impact in an industry that has been dominated by large, vertically integrated vendors.
- It enables a common software solution for call processing to be applied in a number of different kinds of network, including combinations of circuit-based networks and packet voice networks using multiple different packet voice formats and physical transports.
- It allows standardized commodity computing platforms, operating systems and development environments to be leveraged, bringing considerable economies to the development, implementation and processing aspects of telephony software.
- It allows a centralized intelligence in a service provider's voice network remotely to control switching devices located in customer premises, a key requirement for the full exploitation of IP telephony in the future.

Separation between Media Gateway (MG) and Media Gateway Controller (MGC) requires a standardized protocol for communication between the two, and an appropriate standard is now emerging. In the next section, we take a look at the evolution of this protocol.

5.2 Protocols for Media Gateway Control

The softswitch concept has been around for several years, and a number of early implementations exist, some of which have had real operational exposure. So far, softswitch solutions have only been applied in core network switching functions, where the call processing functionality is largely limited to call setup and teardown, and where the complex features required for local telephony are not applicable. Nevertheless, early experience in this field has done much to shape the design of the protocol that MGCs will use to communicate with MGs.

Today's softswitch solutions are mostly based on a protocol called Media Gateway Control Protocol (MGCP), which evolved from two earlier proposals called Simple Gateway Control Protocol (SGCP) and Internet Protocol Device Control (IPDC). MGCP has been published as RFC2705, but its status is Informational and it is not on the standards track.

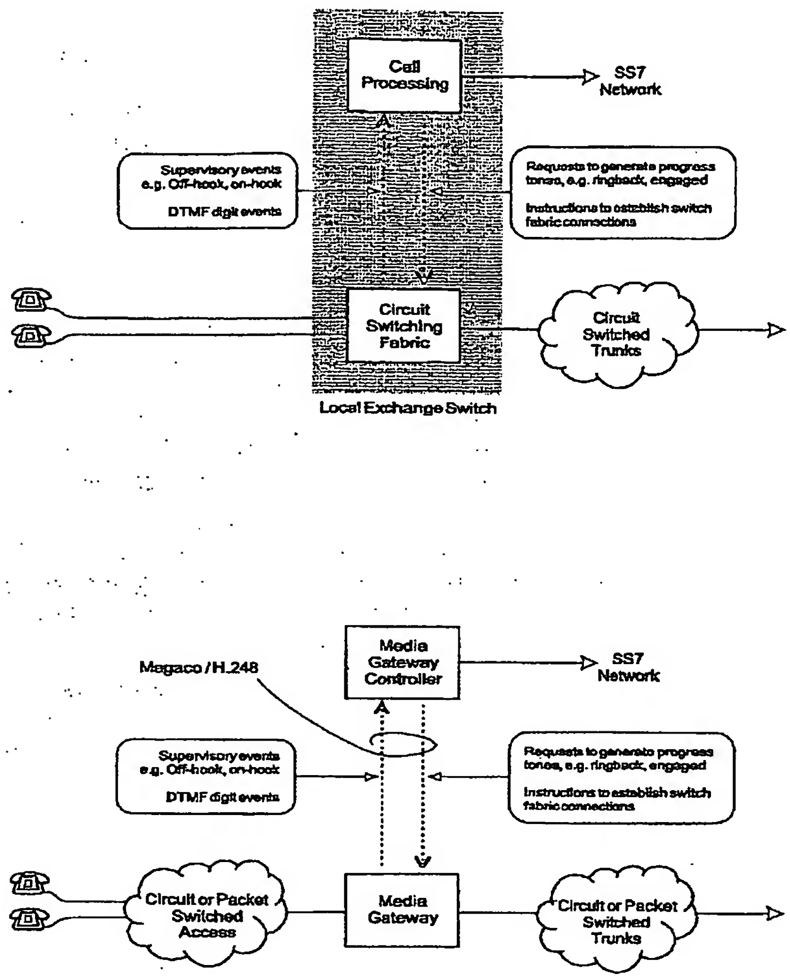

Figure 2 : Comparison between Local Exchange Switch and Softswitch

MGCP is a very IP-centric protocol and does not have effective capabilities to handle other packet voice transports such as voice over ATM. Operational experience with MGCP also identified some practical shortcomings, such as the lack of an effective method for a Media Gateway Controller to obtain information about the capabilities of a Media Gateway. As a result, MGCP itself has largely been abandoned, but work continues within the Internet Engineering Task Force (IETF) in the Media Gateway Control (megaco) working group, on a new protocol known as Megaco. The International Telecommunications Union (ITU) has assigned the identity H.248 to this new protocol, and has adopted the text of Megaco from the IETF work. The two organizations are working together to finalize the Megaco specifications by the end of 2000.

5.3 Capabilities of Megaco

The Megaco protocol provides a comprehensive solution for the control of MGs. As with earlier generations of media gateway control, Megaco is based on the principle that all call processing intelligence resides in the MGC. The MG does not retain knowledge of call state, it provides only the capability to cross-connect various kinds of media streams under the control of the MGC, and to detect and transmit various kinds of signaling associated with those media streams.

Megaco views the MG as a collection of "terminations", each of which represents a certain kind of media stream. A termination may be a fixed physical entity such as an analog line or a DS0 timeslot in a DS1 interface, or it may be a logical entity such as a VoIP packet stream. Logical terminations may be created and destroyed by means of Megaco commands.

Cross-connections within the MG are created by means of Megaco commands that request two or more terminations to be placed in the same "context". If the media streams associated with terminations that are in the same context are of different types (for instance, one is a DS0 timeslot while the other is a VoIP packet stream) then the MG is expected to perform appropriate media conversion between them. To support this, terminations have various media stream properties associated with them such as the identity of the voice encoding that is to be used.

Terminations have other properties such as a list of signaling events that they are expected to notify to the MGC, and a list of signals that they are capable of transmitting on request from the MGC. For example an analog line termination should be capable of notifying the MGC when it sees an off-hook or an on-hook event taking place, and should also be capable of applying ringing on the line when requested by the MGC. The events and signals that are associated with a specific type of termination are described in a "package".

Megaco is designed to be an extensible protocol, and it includes a mechanism to permit the specification and registration of new packages. This extensibility overcomes a major shortcoming of earlier media gateway control protocols such as MGCP, since it addresses the needs of packet voice protocols other than VoIP and provides the means to handle country-specific variations of analog telephony services.

5.4 Transport of Megaco

The Megaco protocol is designed to be transport independent, although the specification does include some appendices that describe the use of both TCP/IP and UDP/IP as transport options. Most softswitch implementations are likely to use an IP-based transport for Megaco, although there may be good reasons to use a native ATM-based transport such as I.366.1 (Segmentation and Re-assembly sublayer for AAL2) to support remote MGs that operate with VoATM connections.

6 Application of Softswitch to Local Telephony

In the previous section, we described the functional components of a softswitch solution. Next we are going to look at the realization of softswitch-based solutions for local telephony. We'll start by identifying various architectural options for the application of softswitch technology in local telephony.

6.1 Network Architecture Requirements

A comprehensive softswitch solution for local telephony requires us to support the following kinds of network connections:

- Access network connections based on traditional telephony technologies, including analog lines, Digital Loop Carrier systems, and digital circuit-based facilities connecting to customer-owned PBXs.
- Access network connections based on packet voice technologies, such as VoDSL.
- Trunk network connections over digital circuit-based facilities to other end-office switches, into local and long distance tandem interconnect switches, and into special facilities such as E911 and operator services tandem switches.
- Trunk network connections to other Media Gateways over packet-based facilities supporting VoIP and/or VoATM.

Each possible combination of access and trunk network type requires a specific combination of Media Gateway capabilities to support local telephony access. The MG functions may be located on the customer premises, in the service provider's network, or both.

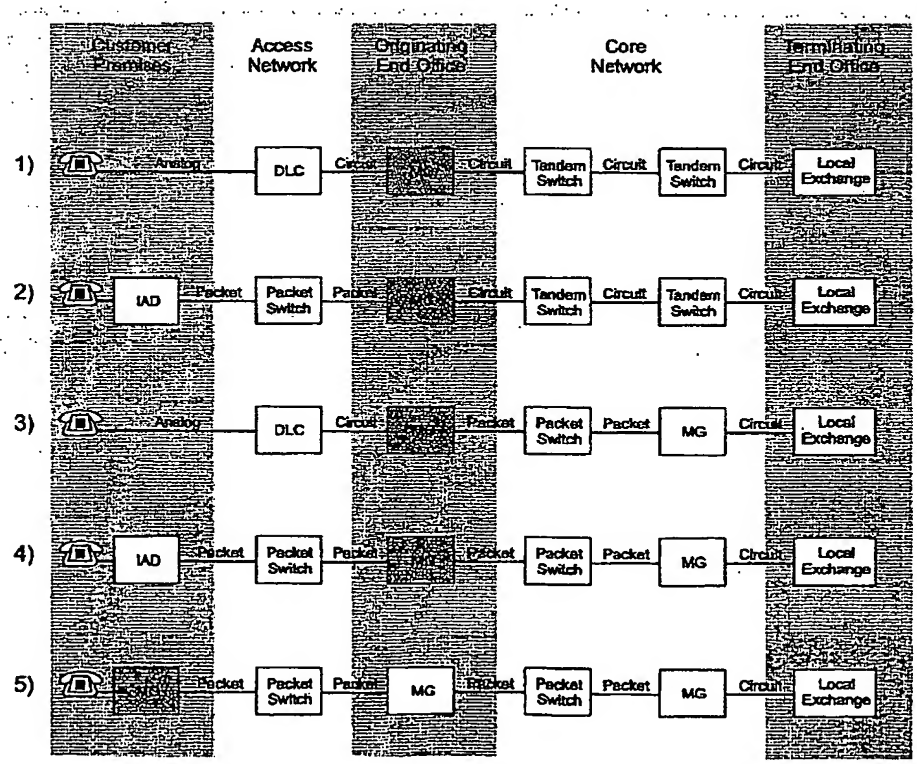

Figure 3 : Network Architectures for Softswitch in Local Telephony

Figure 3 shows five examples of network architectures that use Media Gateways to create softswitch-based solutions for local telephony. This diagram illustrates five network "zones" from customer premises through access network, originating end office, core network and terminating end office. In each of these examples, the terminating end office is shown as a conventional circuit-based local exchange switch, but it could equally well be a Media Gateway as shown in the originating end office. In each example, the MG that provides dial tone and other end office switching functions is shown in gray. Note that for clarity's sake, the Media Gateway Controller that is associated with each Media Gateway is not shown.

Example (1) looks almost exactly like today's phone network, with the exception that the originating end office function is shown here being performed by a MG. In this case, the MG is simply a circuit switch. There is nothing in the definition of a MG that requires it to perform media conversion, and Megaco is fully capable of supporting the control of a pure circuit-switching Media Gateway from a Media Gateway Controller.

In example (2), the conventional circuit-based access network is replaced by a VoDSL network. An Integrated Access Device (IAD) on the customer premises delivers packet voice access. Although the IAD does perform media conversion between analog ports and packet voice streams, it is not a Media Gateway in the softswitch sense, because it is not controlled by an external MGC. The MG in the originating end office performs media conversion between the packet voice access network and the circuit-based core network.

Example (3) shows a conventional telephony access network, but replaces the circuit-based core network with packet voice. The MG in the originating end office performs media conversion between the circuit-based access network and the packet-based core. A second MG performs trunk conversion to enable the call to terminate in a conventional circuit-based local exchange. Note that the softswitch functionality associated with this second MG is far less complex than that associated with the MG in the originating end office, because the trunk conversion MG only has to deal with call setup and teardown, not with any of the special calling features that are handled by the MG in the originating end office.

Example (4) combines the packet-based access network of example (2) with the packet-based core of example (3), to extend the reach of packet voice all the way from the originating customer premises to a Media Gateway that is located as close as possible to the terminating end office.

Example (5) moves the originating end office MG functionality out to the customer premises. The customer premises MG is controlled by a MGC that resides in the public network. Although this architecture is superficially very similar to example (4), there are some major benefits to this approach which are covered in more detail later in this paper.

In the examples that include multiple MGs, each MG may be controlled by a separate MGC, or a single MGC may control two or more of the MGs shown. A collection of MGs that is controlled by a single MGC behaves like a single distributed Media Gateway. Telephony signaling protocols are used to support the switching of calls between MGs that are controlled by different MGCs. Where the network that connects MGs is circuit-based, then conventional telephony protocols such as SS7 may be used between MGCs. Where the network is packet-based, new signaling protocols such as H.323 or Session Initiation Protocol (SIP) are required.

6.2 Media Gateway Controller Requirements

The examples illustrated above identify two distinct categories of Media Gateways: MGs that reside either in the end office or on the customer premises, and which deliver dial tone and support the full range of end office switching functions; and MGs that provide trunk conversion capabilities, needing only to support basic call setup and teardown.

MGs deployed in today's networks are all of the trunk conversion variety. They are primarily deployed to support interconnection of circuit switches over packet trunking facilities, and provide a tandem switch replacement function.

MGs that deliver end office switching capabilities have yet to be demonstrated – perhaps because the MGC functionality that is required to support end office switching is orders of magnitude more complex than that required for simple trunk conversion.

An "end office" or "local exchange" MGC must implement three layers of functionality fully to meet the needs of a comprehensive softswitch-based local telephony solution: a call agent, a set of basic calling features, and a feature creation environment.

6.2.1 Call Agent

Call agent is the term used to describe the basic MGC functionality needed to set up and tear down calls, and to maintain details of the state of each call. The call agent interacts with the signaling protocols that exist on either side of MG for the purposes of coordinating call setup and teardown. For example, an MG that supports the conversion of SS7 Inter-Machine Trunks to VoIP connections requires an MGC with a call agent that can handle SS7 ISUP messages and the H.245 call control messages that support VoIP call establishment as part of the H.323 protocol suite.

Media Gateways that support only trunk conversion need little more than basic call agent functionality in the MGC, and most of today's softswitch solutions comprise just such a call agent and little else. In a softswitch-based solution for local telephony, the call agent is the lowest of three layers of functionality.

The call agent includes call routing functionality. In general, call routing is more complex in a local exchange switch than it is in a tandem switch in the core of the network. This is because the local exchange switch has to support special call routing capabilities including operator services routing, E911 call routing, 800 number translation, Local Number Portability (LNP), Preferred Interexchange Carrier (PIC) code routing, and Carrier Access Code (CAC) routing.

6.2.2 Basic Calling Features

An MGC that contained only call agent functionality could provide local telephony service to customers that use PBXs. This is because PBXs deliver the calling features required by the end user, and the connection from the PBX into the public network requires only basic transport services, which can be provided by a call agent.

Small business and residential customers do not, generally, have their own PBXs. They expect a set of calling features such as caller ID, call waiting, call forwarding, voicemail and so on to be provided by the network. Service providers are making good margins on these services today, and subscribers have become dependent on them. Therefore, they must be provided as part of any softswitch solution for local telephony.

Call Waiting
Call Forward All
Call Forward Busy Line
Call Forward Don't Answer
CFDA Subscriber Ring Control
Remote Activation of Call Forwarding
Three-Way Calling
Call Transfer
Call Pickup
Distinctive Ringing
Do Not Disturb
Calling Number Delivery
Calling Number Delivery Blocking
Caller ID with Call Waiting
Calling Name Delivery
Calling Name Delivery during AC / AR Ringing
Automatic Callback
Automatic Recall
Unidentified Call Rejection
Selective Call Forwarding
Selective Call Rejection
Customer Originated Trace
Business Group Line
Business Group Dialing Plan
Semi-Restricted Line
Fully-Restricted Line
Intercom Dialing
Special Intercept Announcement
Critical Inter-digit Timing for Dialing Plan
Customer Access Treatment Code Restriction
Direct Inward Dialing / Direct Outward Dialing DID/DOD
Automatic Identified Outward Dialing
Single-Digit Dialing
Busy Line Verification
Busy Line InterruptE
E911 Operator Services

Figure 4 : Some Basic Local Telephony Calling Features

6.2.3 Feature Creation Environment

An MGC that just implemented a call agent and a set of basic calling features might provide a softswitch-based solution for local telephony that offers comparable functionality to a traditional local exchange switch – perhaps at substantially lower cost – but it would not provide any means for a service provider to create differentiated services. This is a key requirement for attracting and retaining customers, so an MGC for local telephony services must have the ability for new services to be created and customized by the service provider.

Today's local exchange switches do not offer any facilities for service creation. All the switch's features are implemented in embedded software, and no public interfaces exist at which features can be added to the switch or existing features modified.

Some special features can be implemented outside the switch using standard signaling interfaces such as SS7, and this is the basis for the Advanced Intelligent Network (AIN) concept. But AIN has promised more than it has been

able to deliver – one of the reasons being that many desirable new features require direct interaction with the call state machine, and AIN does not provide that capability.

Developing new features in a traditional local exchange switch requires opening up the embedded software and writing code to the internal APIs of the switch. In most current switches, the software codebase has evolved for many years, and the sheer complexity of the code almost guarantees that complex interactions are involved in any new feature development. Consequently, exhaustive regression testing is required to verify that the addition of a new feature has not introduced errors into any of the rest of the features.

If service providers are to have the opportunity to create their own new features, an entirely new approach is required to the architecture of the feature processing software in the MGC.

- A high level language is required to define the functionality of new features, and this language should be linked to graphical tools that enable new features to be designed visually, without complex coding rules.
- A fully object-oriented approach is required, in which basic feature primitives are implemented as objects, and new features can be built upon these feature primitives, taking advantage of inheritance.
- The object model for call processing should take account of the possible interactions between basic feature primitives, making it unnecessary to carry out extensive regression testing of any new feature that has been created from the basic feature primitives.
- The data that describes each subscriber's configured features and the parameters that control them should be accessible through an open, self-describing data exchange format that facilitates Web-based subscriber self-care. Extensible Markup Language (XML) is ideal for this purpose.

An MGC that takes this approach and offers an easy-to-use and robust feature creation environment places a great deal of power in the hands of telephony service providers. The low cost and rapid development time for new features means that, for the first time, service providers can afford to cater to the specialized needs of specific market segments rather than falling back on the generic features that are found in every switch today. Just a single unique new feature that meets the needs of a particular group or community could attract and retain large numbers of new customers, without the necessity to offer large discounts as an inducement.

6.3 Integration of Softswitch Functionality

The Media Gateway Controller functions described above can be thought of as "layers" in that the call agent, the basic calling features and the feature creation environment add progressively more sophisticated capabilities to the MGC. But these functions should not be treated as layers in the sense of a protocol stack, where each layer communicates with the one above and below by means of well-defined Application Programming Interfaces (APIs), because this approach imposes undesirable restrictions on the capability of the MGC.

The restrictions imposed by a strictly layered approach have become apparent over a number of years of experience with "programmable switches" which implement an embedded call agent and which support well-defined APIs to an external server which implements advanced features. This type of switch is widely used for special telephony applications such as calling card processing. The limitation of the layered approach is that the call state model and its interactions with the external feature server are fixed. A new feature that requires new kinds of call state or a new kind of interaction with the call state model cannot be implemented in this environment.

The concept of separating basic call control from advanced features has found its way into the softswitch world. Many of the softswitch solutions that are now being proposed consist of three elements: a Media Gateway, a Media

Gateway Controller that just handles basic call control, and a "feature server" in which the advanced features and feature creation environment are supported. This model suffers from the same inflexibility as the programmable switch, since the fixed nature of the APIs between the MGC and the feature server make it impossible to exploit new kinds of interaction between features and the call state machine.

A vertically integrated approach to Media Gateway Controller functionality provides a much more future-proof solution. In this model, the call agent, the basic features and the feature creation environment are all running in the same system. This allows us to make the call state machine fully accessible to the feature creation environment, so that we retain complete flexibility to create innovative new features that may require unforeseen call state interactions in the future.

6.4 Operating Environment for the MGC

So far, we have described the MGC function as a body of software without identifying what kind of platform it will be running on. One of the objectives of separating the physical switching function from the call processing function is to permit call processing to run on industry-standard platforms, and leverage low cost processing power and storage, and open operating system and development environments.

In practice, this means that the MGC should run on a high-availability Unix server platform in a dual-redundant configuration. For ultimate protection, the two servers that make up the dual-redundant configuration could be placed in separate physical locations, so long as we ensure that there is sufficient network bandwidth available between them to support the call state replication that is required for complete fault tolerance.

An individual MGC may control a single MG or a collection of MGs, depending on its call processing capacity. Configurations that include multiple MGs controlled by a single MGC behave like a single distributed MG when viewed from the outside.

6.5 Signaling Connections for the MGC

MGCs that control Media Gateways connected to circuit-based or packet-based voice networks must support the termination and processing of the telephony signaling protocols associated with those networks.

For packet-based voice networks, the signaling protocols most likely to be used include H.245 (the signaling protocol that belongs to the H.323 protocol suite) and Session Initiation Protocol (SIP). These protocols ride on top of an IP transport, and since the MGC is likely to have an IP network connection for the transport of Megaco, this connection can be shared for the transport of the packet voice signaling.

For circuit-based networks, the applicable signaling protocols include SS7, ISDN Primary Rate Interface (PRI) signaling, and Channel Associated Signaling (CAS). These protocols are usually transported on circuit-based facilities. In the case of message-based protocols such as SS7 and PRI, the messages are carried on a connection-oriented data link layer such as LAPD (Link Access Protocol for D-channels) or SS7 MTP2 (Message Transfer Part 2), which in turn occupies one or more DS0 timeslots on a T1 facility. CAS is a bit-oriented protocol that is carried in the framing information on T1 facilities.

The MGC function does not typically contain the physical interfaces needed to connect to and terminate these circuit-based signaling protocols. Since the MGC's native mode of communication with the outside world is IP-based, the obvious solution is to map each of the circuit-based signaling protocols onto an IP-based transport.

This is the purpose of SIGTRAN, a proposal being developed in the IETF for a method of carrying circuit-based signaling protocols over IP networks. The key component of this solution is the Simple Control Transmission Protocol (SCTP), a protocol that can efficiently carry multiple sessions of various signaling protocols over a single IP connection.

Since circuit-based signaling protocols are typically associated with and physically carried on the circuit-based transmission facilities that terminate on Media Gateways, the obvious solution for these protocols is to implement a SIGTRAN signaling gateway as an additional function within the MG. This signaling gateway will terminate the lower layers of the SS7 and PRI protocols, and encapsulate the application layer messages for forwarding over the IP connection to the MGC.

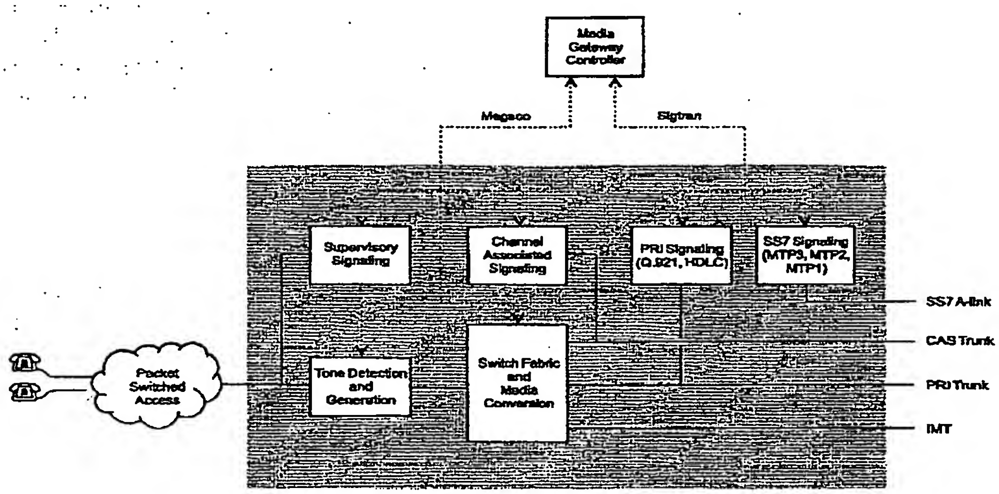

Figure 5 : Media Gateway for Packet-to-Circuit Access with Integrated Signaling Gateway

7 Migration Path for Packet Voice in Local Telephony

We have described an approach to delivering local telephony services with a softswitch-based solution that can lower the cost of local telephony switching infrastructure, deliver differentiated services, and ease the migration to end-to-end packet voice. In this section, we explore how today's VoDSL solutions, which offer a packet voice access network connecting to a conventional local exchange switch, can provide the foundation for migrating to softswitch-based local telephony.

7.1 VoDSL Today

Today's solutions for VoDSL leverage the broadband access network to provide transport for multiple lines of telephony over a single broadband connection. These VoDSL solutions invariably consist of two elements: an Integrated Access Device (IAD) located in the customer premises which packetizes voice and voiceband data transmissions, and an Access Gateway located in the service provider's network that converts packetized voice back into a circuit-based format suitable for delivery to a local exchange switch.

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In this approach, there is no switching of voice traffic within the broadband network. The first switching operation that voice traffic experiences within the service provider's network takes place at the local exchange switch that is upstream of the Access Gateway. Although an individual IAD may have packet connections to multiple Gateways, each port of an IAD must logically be tied to a specific local exchange switch. This fixed relationship between IAD ports and local exchange switch ports is essential to the proper delivery of local telephone service, and has given rise to the term "Loop Emulation" to describe this class of solution.

7.2 Transformation of the Access Gateway

If the VoDSL Access Gateway has been designed with sufficient architectural flexibility, then it should be possible to add support to it for media gateway control via Megaco. In this case, the Access Gateway becomes a true Media Gateway, and if it is paired up with a suitable Media Gateway Controller that supports local telephony features, then it should be able to deliver local telephone service over the VoDSL connection without the need for a conventional local exchange switch.

7.2.1 Media Gateway Supporting Circuit Trunks

This transformation of an Access Gateway into a Media Gateway can be accomplished without any change to the packet voice protocol that is used between the IADs and the Gateway. The new capabilities offered by the Media Gateway are transparent to the IADs, which are unaware that dial tone is now originating from the Media Gateway to which they are connected, instead of a local exchange switch that lies upstream of the Gateway.

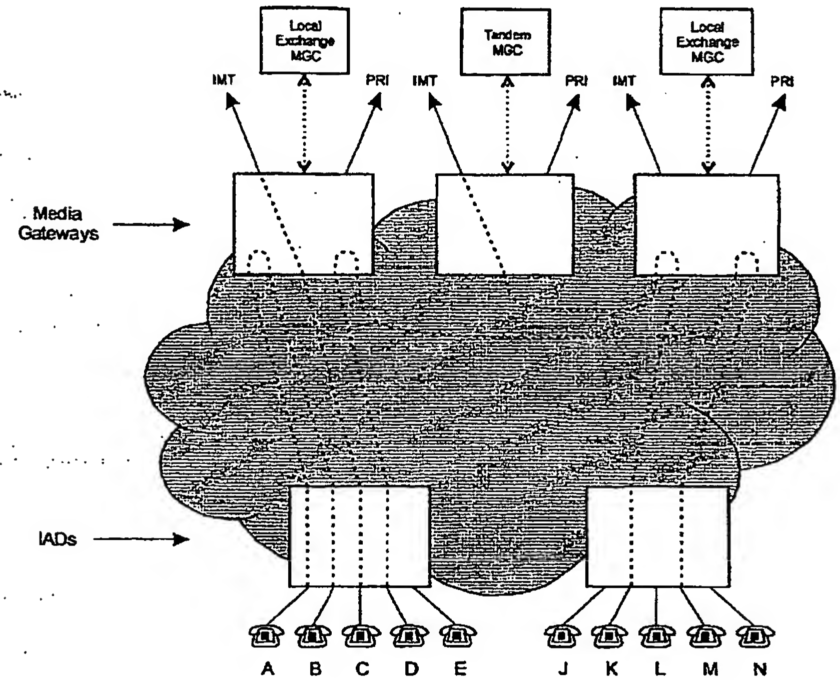
To function as a Media Gateway that supports existing VoDSL IADs, the Gateway requires some additional signal processing hardware resources to perform the following:

- Generation of call progress tones, such as dial tone, ringback, busy, fast busy, stutter dial tone etc.
- Generation of voiceband data transmission tones such as those needed for sending Caller ID information to the user.
- Detection and collection of DTMF dial digits.
- The Access Gateway is connected into the PSTN via circuit-based connections that support an access network protocol such as GR-303 or V5.2. The transformed Media Gateway can make use of the same physical interfaces, but these are now connected to tandem switches and are controlled by trunking protocols such as SS7 and PRI. Ideally, the Media Gateway will have the ability to terminate the physical, data link and network layers of these signaling protocols, passing the signaling application layer messages to the Media Gateway Controller over an IP connection using a standard transport encapsulation such as SIGTRAN.

7.2.2 Media Gateway Supporting Packet Trunks

The Media Gateway described in the previous section supports packet voice access (such as VoDSL) with conversion to circuit-based trunks. This type of functionality is an absolute requirement for a local telephony softswitch solution, since much of the traffic that originates at an IAD will terminate in the same local calling area on a regular POTS connection served by a conventional circuit-based local exchange switch. Therefore most of the packet traffic arriving at the Media Gateway on the access side will have to be handed off to a local access tandem switch via a circuit-switched trunk, typically an SS7 IMT (Inter-Machine Trunk).

It will not be long, however, before packet-based voice trunking is required in this scenario. The packet trunks may be used to network together multiple Media Gateways in the same local calling area to provide a kind of distributed local switch solution, or they may be used to hand off voice traffic to long distance packet voice service providers.



Case 1: Phone C makes a local call with handoff to another service provider
Case 2: Phone K makes a call via a remote tandem gateway with handoff to another service provider
Case 3: Phone A makes a station-to-station call to Phone B
Case 4: Phone D makes a call to Phone M which is on another IAD

Figure 6 : Use Cases for Packet to Circuit Access from IADs

In either case, the Media Gateway will need to be able to switch packet voice traffic arriving from the access side onto outgoing packet voice trunks. If the access network and the trunks are using the same packet voice technology, for example ATM AAL2, then the Media Gateway acts simply as a packet switch to move voice packets between the access and trunk networks. If different packet voice technologies are used on either side of the Media Gateway, then the MG will have to perform packet voice protocol conversion. This can be done without having to convert the voice back to TDM form and re-packetize. An AAL2 voice packet can be converted to a VoIP voice packet by mapping the AAL2 packet header to an IP/UDP/RTP packet header.

For a VoDSL Access Gateway successfully to migrate to this role of packet voice switching, it must have a high performance and fault-tolerant packet-switching bus. If the Access Gateway is designed to convert incoming packet voice traffic to a TDM bus, then it will not be able to handle the passing through of packet voice traffic without introducing an additional cycle of de-packetization and re-packetization. This is undesirable because it adds substantially to transmission delay, and potentially squeezes end-to-end delay budgets that are already tight.

7.3 Migrating the Media Gateway to the Network Edge

The transformation of an Access Gateway into a Media Gateway that we have just described brings us most of the advantages of the softswitch approach without making any changes whatsoever to the IADs that provide the customer connections into the packet access network. The combination of Media Gateway and Media Gateway Controller will allow service providers to deliver local telephony services that leverage the economic benefits of packet voice access, at a small fraction of the cost of a typical local exchange switch.

The economics associated with this kind of solution are such that CLECs can profitably serve local markets where they may only expect to win a few thousand lines. They will be able to attract customers by combining bundled voice and data services with innovative and narrowly targeted new voice features, instead of relying on deep discounts on voice services as they did in the past.

But there is one further optimization that may be useful for certain types of customers. The solution we have described so far relies on a Media Gateway located in the service provider's network to provide dial tone, and this MG acts effectively as an end office switch. As a result, each IAD voice port must be permanently tied to a specific MG in the service provider's network, since the MG represents a single point of ownership for the line in handling features like call waiting and call forwarding.

A consequence of this is that all traffic from a given IAD port has to physically pass through the MG that owns it. This is not necessarily a bad thing; the MG provides essential media conversion functions on most calls to provide the hand-off to local and long distance circuit-switched connections, and to perform packet voice protocol conversion where necessary between packet access and trunk networks.

There are three circumstances in which may be undesirable for all IAD-originated traffic to pass through the same MG in the service provider's network, and we will explore these below. But first, let's look at a solution that de-couples the IAD from a fixed connection with a given service provider's MG.

7.3.1 The IAD as Media Gateway

The functional model we have assumed so far assumes no switching takes place in the network between the IAD and the service provider's MG. Each IAD port is logically "hard-wired" to an MG in the service provider's network. This is necessary if dial tone is being originated in the service provider's MG, since the MG has to have exclusive control of the IADs ports in order successfully to handle complex features such as call waiting and call forwarding.

It follows that, to move away from this model, we have to provide dial tone in the IAD itself. We can do this by moving Media Gateway functionality into the IAD, and by establishing a Megaco connection with the Media Gateway Controller. The MGC must still reside in the service provider's network, since it must connect into the SS7 network to support calls over the PSTN, and it is not feasible for small business or residential users to have their own MGCs with SS7 connections.

We have already described the additional functionality necessary to transform an Access Gateway into a Media Gateway. It comprises mainly signal processing hardware to perform tone generation and recognition, as well as

support for the Megaco protocol itself. By adding these capabilities to the IAD, we can transform it into a Media Gateway that brings end office switching functionality to the customer premises.

The IAD/MG will support analog telephony user ports and a packet voice access port, typically over a DSL connection. Unlike the IAD we have previously described, the IAD/MG does not have a fixed association with a service provider's MG. It is free to establish packet voice connections with any device on the packet network that can support the same protocols. This freedom is, of course, strictly under the control of the MGC that is in the service provider's network. The MGC sees all call setup activity and is able to track all calls by calling number, called number, time of day and duration. With this information, the MGC can produce the call detail records that are necessary for billing purposes.

The service provider's MGC is responsible for analyzing the dial digits collected by the IAD/MG and determining how to route the call. If, say, a call is destined to terminate on another service provider's local exchange switch in the same calling area, the MGC will direct the IAD/MG to establish a packet connection with the appropriate MG in the service provider's network where the packet-to-circuit conversion can be performed and the call can be directed to the appropriate local access tandem.

For this model to operate successfully, it must be possible for the IAD/MG to establish packet voice connections to arbitrary Media Gateways on request. If the packet voice access is based on VoIP, the MGC will supply the IAD/MG with the IP address of the target MG, and instruct it to set up a VoIP session. If the packet voice access is based on VoATM, then the MGC will instruct the IAD/MG to establish a Switched Virtual Circuit to the target MG, and provide the relevant ATM address.

Having described a solution that dissolves the fixed relationship between an IAD and a service provider's MG, let us now explore the circumstances in which this solution delivers real value.

7.3.2 Direct Packet Voice Connections Between IADs

A business customer may have multiple locations that are served by packet voice access, where the IADs are all connected to a common packet network. If there were a substantial amount of intra-enterprise voice traffic between and among these IADs, it would be more efficient for this traffic to travel on direct packet connections between IADs rather that via service provider MGs.

A service provider may choose to support this kind of scenario with a virtual private voice network solution. In this case, the service provider's MGC could be configured to handle calls among this community of IAD/MGs using, say, a 4-digit dial plan. The community of IADs would then behave like a distributed PBX. While VPN is not a new idea, this use of packet voice to implement a VPN solution under the control of a service provider's MGC brings the concept right up-to-date.

7.3.3 Direct Packet Voice Connections to Remote Trunking Gateways

As packet voice networking becomes ubiquitous, it should be possible for an IAD/MG to make direct packet voice connections with remote gateways that perhaps lie on the far side of a long distance or international packet voice network. If the packet voice access network and the long distance packet voice network are operated by different service providers, this may require packet voice peering agreements to be in place. The IAD/MG and the remote gateway must use the same packet voice transport protocol in order to be able to interoperate in this way. For this reason, support for multiple different packet voice protocols (VoIP and VoATM) is desirable in the IAD/MG.

7.3.4 Station-to-station Calls at the IAD

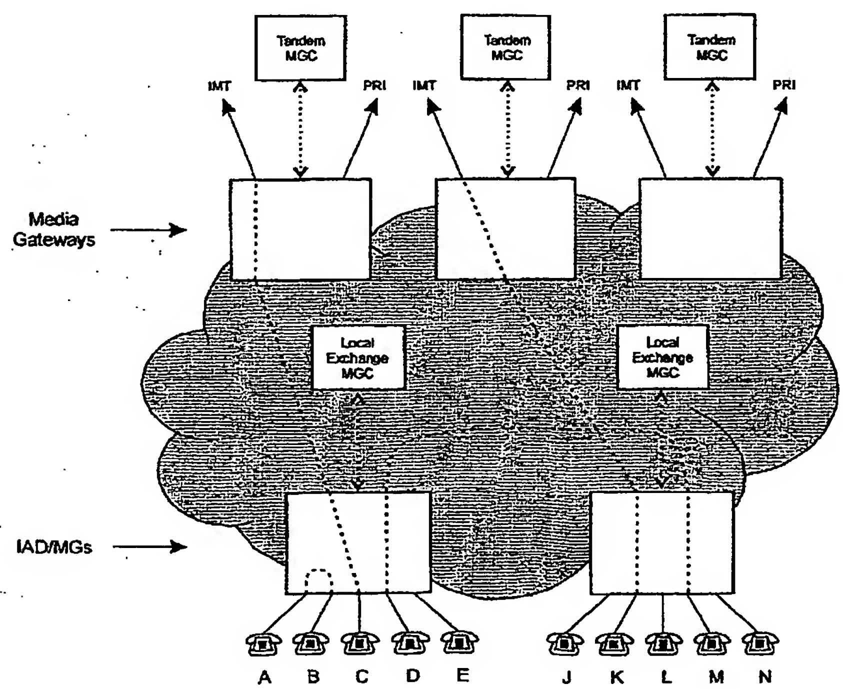
Many businesses require support for local station-to-station calling, and today they have a choice of using local switching based on a PBX or key system, or network switching using Centrex-type services.

PBXs and key systems can be used with packet voice access networks, but the access network connections have to be configured in specific ways to deal with the presence of local switching. Typically, a "hunt group" is configured for outgoing calls and for calls to the main switchboard or auto-attendant number, while an additional Direct Inward Dialing (DID) group must be configured to handle direct dialed incoming calls. With this setup, all the required local calling features such as call waiting must be implemented in the PBX.

Not all businesses are prepared to deal with the cost and complexity of PBXs. The alternative solution has been to use Centrex, where the local exchange switch provides support for the 4 or 5 digit dial plan that is used for station-to-station calling, as well as offering access to outside lines and direct inward dialing.

One of the problems with Centrex is that all station-to-station calls physically pass through the local exchange switch, and therefore consume resources in the local access network. A Centrex customer with 100 stations requires 100 voice connections to the local exchange.

Using an IAD/MG with local switching support, a service provider could offer a kind of "next generation" Centrex service where all the call handling and feature processing intelligence belongs in the MGC that is part of the service provider's network, but local switching is performed directly in the IAD/MG. The customer would gain all of the utility benefits of Centrex while the service provider would only have to provide sufficient access network resources to support external calls. For example, a typical installation with 100 stations might need only 16 outside line connections, and these could be delivered via packet voice over a single DSL connection.



Case 1: Phone C makes a local call with handoff to another service provider
Case 2: Phone K makes a call via a remote tandem gateway with handoff to another service provider
Case 3: Phone A makes a station-to-station call to Phone B
Case 4: Phone D makes a call to Phone M which is on another IAD

Figure 7 : Use Cases for Packet to Circuit Access from IAD/MGs

8 Conclusion

In this paper, we have described a migration path for broadband packet voice access from a transport-only solution that relies on a conventional local exchange switch to a fully-fledged switching and access solution that delivers "packet voice dial-tone."

By far the most important ingredient in this migration path is the softswitch technology that forms the basis of the Media Gateway Controller. The softswitch is where all the service intelligence resides for the delivery of local telephone services. A far higher level of capability is required for this type of softswitch than for today's tandem softswitch applications where functionality is limited to interworking between PRI, SS7 and VoIP signaling. For local telephony, the softswitch needs not only to deliver a critical mass of local telephony features, but also to provide a feature creation environment that enables service providers to develop differentiated services.

The development of a softswitch that is truly capable of meeting these requirements is a major undertaking. But for those who succeed in reaching this goal, an awesome prospect lies ahead: nothing less than the total transformation of the local telephone network.

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